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Title: SYSTEM AND METHOD FOR PROVIDING A WIRELESS DATA NETWORK TELEPHONE SYSTEM

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SYSTEM AND METHOD FOR PROVIDING A WIRELESS DATA NETWORK TELEPHONE SYSTEM

BACKGROUND OF THE INVENTION

A. Field of the Invention

The present invention is related to a method and system for providing communication services over a network. In particular, the present invention relates to a system and method for delivering telephone services to residential users using a wireless network.

B. Description of the Related Art

For many years, telephone service providers on the Public Switched Telephone Network (PSTN) provided their customers nothing more than a telephone line to use to communicate with other subscribers. Over time, telephone service providers have enhanced their service by providing Custom Local Area Signaling Service (CLASS) features to their customers. Similar communication services are provided by a Private Branch Exchange (PBX), which is typically implemented in a nonresidential setting.

The CLASS features permit customer subscribers of the features to tailor their telephone service according to individual needs. Some of the more well-known CLASS features are:

- Call blocking: The customer may specify one or more numbers from which he or she does not want to receive calls. A blocked caller will hear a rejection message, while the callee will not receive any indication of the call.

- Call return: Returns a call to the most recent caller. If the most recent caller is busy, the returned call may be queued until it can be completed.

- Call trace: Allows a customer to trigger a trace of the number of the most recent caller.

- Caller ID: The caller's number is automatically displayed during the silence period after the first ring. This feature requires the customer's line to be equipped with a device to read and display the out-of-band signal containing the number.

• Caller ID blocking: Allows a caller to block the display of their number in a callee's caller ID device.

• Priority ringing: Allows a customer to specify a list of numbers for which, when the customer is called by one of the numbers, the customer will hear a distinctive ring.

• Call forwarding: A customer may cause incoming calls to be automatically forwarded to another number for a period of time.

A customer subscriber to a CLASS feature may typically activate and/or de-activate a CLASS feature using "*" directives (e.g., *69 to automatically return a call to the most recent caller). CLASS features may also be implemented with the use of out-of-band data. CLASS feature data is typically transmitted between local Class-5 switches using the Signaling System 7 (SS7).

Local Exchange Carriers (LECs) and other similar organizations maintain CLASS offices that typically contain a database entry for each customer. The database allows specification of the CLASS features a customer has subscribed to, as well as information, such as lists of phone numbers, associated with those features. In some cases, customers may edit these lists on-line via a touch-tone interface. A list of all phone numbers that have originated or terminated a call with each customer is often included in the CLASS office database. For each customer, usually only the most recent number on this list is stored by the local Class-5 switch.

A Private Branch Exchange (PBX), is a stored program switch similar to a Class-5 switch. It is usually used within a medium-to-large-sized business for employee telephony service. Since a PBX is typically operated by a single private organization, there exists a wide variety of PBX services and features. Custom configurations are common, such as integration with intercom and voice mail systems. PBX's typically support their own versions of the CLASS features, as well as other features in addition to those of CLASS. Most PBX features are designed to facilitate business and group communications.

A summary of typical PBX features includes:

• Call transfer: An established call may be transferred from one number to another number on the same PBX.

• Call forwarding: In addition to CLASS call forwarding, a PBX number can be programmed to automatically transfer a call to another number when the first number does not answer or is busy.

• Camp-on queuing: Similar to PSTN call return, a call to a busy number can be queued until the callee can accept it. The caller can hang up their phone and the PBX will ring them when the callee answers.

• Conference calling: Two or more parties can be connected to one another by dialing into a conference bridge number.

• Call parking: An established call at one number can be put on hold and then reestablished from another number. This is useful when call transfer is not warranted.

• Executive override: A privileged individual can break into an established call. After a warning tone to the two participants, the call becomes a three-way call.

While the CLASS and PBX features have enhanced the offerings of service providers that use the PSTN, the features are nevertheless limited in their flexibility and scope. The effect to the user is that the features become clumsy and difficult to use. For example, in order to use the Call Forwarding function, the user must perform the steps at the user's own phone prior to moving to the location of the telephone to which calls will be forwarded. A more desirable approach, from the standpoint of usefulness to the user, would be to perform the steps at the telephone to which calls will be forwarded.

Much of the lack of flexibility of the PSTN features is due to the lack of flexibility in the PSTN system itself. One problem with the PSTN is that the terminal devices (e.g. telephones) lack intelligence and operate as "dumb" terminals on a network having the intelligence in central offices. Most PSTN telephones are limited in functional capability to converting the analog signals they receive to sound and converting the sound from the handset to analog signals.

Some PSTN telephones have a display device and a display function to display specific information communicated from intelligent agents in the PSTN network using the PSTN signaling architecture. For example, some PSTN telephones have a display function to enable the Caller ID feature. Even such PSTN telephones are limited however by the closed PSTN signaling architecture, which prohibits access by the PSTN

telephones to the network signaling protocols. The display functions are effectively limited to displaying text, again, as a "dumb" terminal.

The Internet presents a possible solution for distributing intelligence to telephony terminal devices. In Internet telephony, digitized voice is treated as data and transmitted across a digital data network between a telephone calls' participants. One form of Internet telephony uses a telephony gateway/terminal where IP telephony calls are terminated on the network. PSTN telephones are connected by a subscriber line to the gateway/terminal at the local exchange, or at the nearest central office. This form of Internet telephony provides substantial cost savings for users. Because the PSTN portion used in Internet telephony calls is limited to the local lines on each end of the call, long distance calls may be made for essentially the cost of a local call. Notwithstanding the costs savings provided by this form of Internet telephony, it is no more flexible than the PSTN with respect to providing enhancements and features to the basic telephone service.

In another form of Internet telephony, telephones are connected to access networks that access the Internet using a router. The telephones in this form of Internet telephony may be substantially more intelligent than typical PSTN telephones. For example, such a telephone may include substantially the computer resources of a typical personal computer.

It would be desirable to incorporate CLASS and PBX features into a data network telephony system that uses a data network such as the Internet.

It would be desirable to provide new features and enhancements to telephony service that accommodates and conforms to users' needs.

It would also be desirable to provide features and capabilities to telephone service that create new opportunities for users and for service providers.

Such opportunities include providing residential telephone service in a residential telephone network. It would be desirable to provide affordable multiple line and multiple number home telephony service.

The present invention addresses the above needs by providing a system in a data network telephony system, such as for example, the Internet, for providing residential telephone service using a wireless network. The residential telephone service allows for

multi-line and multi-number service that is easily configured. The residential telephony system also enables users to connect to Internet services using a portable information device.

BRIEF DESCRIPTION OF THE DRAWINGS

Presently preferred embodiments of the invention are described below in conjunction with the appended drawing figures, wherein like reference numerals refer to like elements in the various figures, and wherein:

FIG. 1 is block diagram of a residential telephone network in a data network telephony system according to one embodiment of the present invention;

FIG. 2 is a block diagram showing a system for using a portable information device (PID) to connect to Internet services on a residential telephone network according to an exemplary embodiment of the present invention;

FIG. 3A is a block diagram of a data network telephone according to an exemplary embodiment of the present invention;

FIG. 3B is a block diagram of the telecommunications base station in FIG. 1 according to an exemplary embodiment of the present invention;

FIG. 3C is a block diagram of a wireless data network telephone according to an exemplary embodiment of the present invention;

FIG. 4 is a block diagram of a PID according to an exemplary embodiment of the present invention;

FIG. 5 is a stack layer diagram showing the layers of an IrDA stack;

FIG. 6 is a block and stack layer diagram illustrating an embodiment of the protocol stacks in an exemplary embodiment of a PID linked to a data network telephone;

FIG. 7A is block and stack layer diagram illustrating an embodiment of the present invention in which a connection to an Internet service may be established;

FIG. 7B is a block and stack layer diagram illustrating an alternative embodiment of the present invention in which a connection to an Internet service may be established; and

FIG. 8 is a combined block and pictorial diagram showing advantageous use of a system for providing PID data exchange according to one embodiment of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

The following patent applications owned by the assignee of the present application are incorporated be reference:

- U.S. Patent App. Ser. No. 09/406,321 "System and Method for Controlling Telephone Service Using a Wireless Personal Information Device" to Schuster, et al., Attorney Docket No. 99,365;
- U.S. Patent App. Ser. No. 09/406,320 "System and Method for Advertising Using Data Network Telephone Connections" to Schuster et al., Attorney Docket No. 99,373;
- U.S. Patent App. Ser. No. 09/405,283 "System and Method for Providing User-Configured Telephone Service in a Data Network Telephony System" to Sidhu, et al., Attorney Docket No. 99,411;
- U.S. Patent App. Ser. No. 09/406,322 "System and Method for Accessing a Network Server Using a Portable Information Device Through a Network Based Telecommunication System" to Schuster, et al., Attorney Docket No. 99,593;
- U.S. Patent App. Ser. No. 09/406,152 "System and Method for Interconnecting Portable Information Devices Through a Network Based Telecommunication System" to Schuster, et al., Attorney Docket No. 99,594;
- U.S. Patent App. Ser. No. 09/405,981 "System and Method for Enabling Encryption on a Telephony Network" to Schuster, et al., Attorney Docket No. 99,595;
- U.S. Patent App. Ser. No. 09/406,128 "System and Method for Using a Portable Information Device to Establish a Conference Call on a Telephony Network" to Schuster, et al., Attorney Docket No. 99,596;
- U.S. Patent App. Ser. No. 09/406,151 "System and Method for Associating Notes with a Portable Information Device on a Network Telephony Call" to Schuster, et al., Attorney Docket No. 99,600;
- U.S. Patent App. Ser. No. 09/406,298 "System and Method for Providing Shared Workspace Services Over a Telephony Network" to Schuster, et al., Attorney Docket No. 99,601;
- U.S. Patent App. Ser. No. 09/406,066 "System and Method for Providing Service Provider Configurations for Telephones in a Data Network Telephony System" to Schuster, et al., Attorney Docket No. 99,602;
- U.S. Patent App. Ser. No. 09/451,388"System and Method for Providing User Mobility Services on a Telephony Network" to Schuster, et al., Attorney Docket No. 99,229;

- U.S. Patent App. Ser. No. 09/470,879 "System and Method for Providing Call-Handling Services on a Telephony Network" to Schuster, et al., Attorney Docket No. 99,914;

- U.S. Patent App. Ser. No. 09/181,431 "Method Apparatus and Communication System for Companion Information and Network Appliances" to Wang, et al.;

- U.S. Patent App. Ser. No. 09/321,941 "Multiple ISP Support for Data Over Cable Networks" to Ali Akgun, et al., Attorney Docket No. 98,638;

- U.S. Patent App. Ser. No. 09/218,793 "Method and System for Provisioning Network Addresses in a Data-Over-Cable System" to Ali Akgun, et al., Attorney Docket No. 99,678; and

- U.S. Patent App. Ser. No. 08/887,313 "Network Access Methods, Including Direct Wireless to Internet Access" to Yingchun Xu, et al., Attorney Docket No. 97,181.

The following patent applications owned by the assignee of the present application and filed concurrently with the application herewith, are incorporated by reference:

- "System And Method For Providing Telephone Service Having Private Branch Exchange Features In A Data Network Telephony System" to Schuster et al., Attorney Docket No. 99,366.

- "System And Method For Providing A Wireless Data Network Telephone System" to Schuster et al., Attorney Docket No. 99,590.

- "System And Method For Accessing A Network Server Using A Portable Information Devices Through A Network Based Telecommunication System" to Schuster et al., Attorney Docket No. 99,592.

- "System And Method For Accessing Radio Programs Using A Data Network Telephone In A Network Based Telecommunication System" to Schuster et al., Attorney Docket No. 99,742.

- "System And Method For Providing Local Information In A Data Network Telephony System" to Schuster et al., Attorney Docket No. 99,838.

- "System And Method For Enabling A Portable Information Device For Use In A Data Network Telephone System" to Schuster et al., Attorney Docket No. 99,741.

- "Dialing Token For Initiating A Telephone Connection In A Data Network Telephone System" to Schuster et al., Attorney Docket No. 99,375.

- "Flexible Dial Plan for a Data Network Telephony System" to Schuster, et al., Attorney Docket No. 99,374.

- "Personalized Call Announcement on a Data Network Telephony System" to Schuster, et al., Attorney Docket No. 99,597.
- "Personalizing a Data Network Appliance on a Data Network Telephony System" to Schuster, et al., Attorney Docket No. 99,598.
- "Proximity-Based Registration on a Data Network Telephony System" to Schuster, et al., Attorney Docket No. 99,599.

The following additional references are also incorporated by reference herein:

- "Multiple ISP Support for Data Over Cable Networks" to Ali Akgun, et al.
- U.S. Patent App. Ser. No. 09/218,793 "Method and System for Provisioning Network Addresses in a Data-Over-Cable System" to Ali Akgun, et al., Attorney Docket No. 98,678

A. PID-Enabled Data Network Telephony System

FIG. 1 is a block diagram showing an exemplary embodiment of a system 100 for providing telephony services and for connecting to Internet services according to one embodiment of the present invention. A wireless residential telephone system 10 is linked to a first access network 112 via connection 130 may communicate over a data network 106 by connecting via the first access network 112. A data network telephone 108 is linked to a second access network 114 through connection 136 and may communicate over the data network 106 by connecting via the second access network 114.

The wireless residential telephone system 10 advantageously connects to the data network 106 for telephony services in a data network telephone system. The wireless residential telephone system 10 in FIG. 1 includes a telecommunications network access station 110, a first data network teleport 18a, a second data network teleport 18b, a third data network teleport 18c, a first portable information device (PID) 110a, and a second PID 110b. Only three data network teleports 18 et seq. are shown in FIG. 1, however, the wireless residential telephone system 10 may be configured to have any number of data network teleports 18 et seq. The data network teleports 18 et seq. are primarily used in a residence as a voice communications device, similar to a telephone in plain-old telephone service (POTS). Telephone service is provided by the data network telephony system 100 described below. The wireless residential telephone system 10 accesses the data

network telephony system 100 using a telecommunications network access station 107, which advantageously provides multi-line and muli-number service to residents.

One of ordinary skill in the art will appreciate that while applications of embodiments of the present invention are described in the context of residential services, embodiments of the present invention are not limited in such a manner. The scope of the present invention is broad enough to encompass wireless telephony systems regardless of the type of application.

The data network 106 in the system 100 typically includes one or more Local Area Networks (LANs) connected to one another or to a Wide-Area Network (WAN), such as an Internet Protocol (IP) network, to provide wide-scale data connectivity. The data network 106 may use Voice Over Packet (VOP) schemes in which voice signals are carried in data packets. The network 106 may also include a connection to the Public Switched Telephone Network (PSTN) to allow for voice connections using traditional circuit switching techniques. In one embodiment, the data network 106 may include one or more LANs such as Ethernet LANs and support data transport protocols for performing Voice-over-Internet-Protocol (VoIP) techniques on the Internet. For further details regarding VoIP, see the information available through the Internet Engineering Task Force (IETF) at www.ietf.org. In addition, an Internet Telephony gateway may be included within the system 100 to allow for voice connections to users connected by subscriber lines at a PSTN Central Office.

The data network 106 may be used to access a variety of Internet services 101. For example, the Internet includes the World-Wide Web102, which is a well-known system for exchanging data over the Internet. The World-Wide Web 102 is commonly used to access targeted information using a computer workstation and an application on the workstation called a browser. With respect to PID's, many Internet Content Providers offer a variety of Web PID data service 103 to permit viewing World-Wide Web data on a PID which is smaller than a workstation. One advantage of the system 100 in FIG. 1 is that Web PID data service 103 and other Internet services 101 may be accessed using the PID 110a,b using a data communication connection. For example, Palm.net offers such services, called Web clippings™. The Wireless Applications

Protocol (WAP) may also be used to implement downloads of information from such services such WAP.Yahoo.DE.

The data network telephone 108 (described further below with reference to FIG. 3) typically include a voice input, a voice output and a voice processing system. The voice processing system converts voice sound to digital data signals that are communicated on a voice connection over the data network. The voice processing system also converts digital data signals received from the voice connection to voice sound. The data network telephone 108 typically include a central processing unit and memory to store and process computer programs. Additionally, each data network telephone 108 typically includes a unique network address, such as an IP address, in memory to uniquely identify it to the data network 106 and to permit data packets to be routed to the device.

The telecommunications network access station 107 includes a data network interface and a unique network address to provide data communications capabilities similar to the data network telephone. The telecommunications network access station 107, however, establishes voice connections over the data network 106 that are communicated to one or more of the data network teleports 18 et seq. The telecommunications network access station 107 includes user account information and device identifiers for each data network teleport 18 et seq. in the wireless residential telephony network 10. The telecommunications network access station 107 includes a station transceiver interface 14, such as a radio-frequency antenna, to perform wireless communication with the data network teleports 18 et seq. One advantage of the system in FIG. 1 is that the data network teleports 18 et seq. do not require their own IP addresses in order to communicate over the data network 106.

The data network teleports 18a-c include a wireless transceiver interface 12, a display 116, a display 116, a keypad 118, a voice input, a voice output and a voice processing system. The voice processing system converts voice sound to digital data signals that are communicated by a wireless connection 20 to the telecommunications network access station 107. The telecommunications network access station 107 communications the digital data signals on a voice connection over the data network 106. The voice processing system also converts digital data signals received from the voice

connection to voice sound. The data network teleports 18a-c may include a central processing unit and memory to store and process computer programs.

In one alternative embodiment, the voice processing system sends voice signals as an analog signal by modulating the wireless signal, using for example, frequency modulation, or other suitable modulation schemes.

A first PID 110a, linked to the first data network teleport 18a via connection 109a may communicate over the data network 106 by connecting via the wireless connection 20 to the telecommunications network access station 107. The telecommunications network access station 107 connects to the first access network 112. A second PID 110b linked to the second data network telephone 108 via connection 109b may communicate over the data network 106 by connecting via the second access network 114. The PIDs 110a-b each contain user attributes stored in a user information data base. The user attributes may contain such information as a user identifier, schedule information, and other information that is associated with a user of the PID 110a or 110b. The PIDS 110a-b each include a user interface allowing a user to easily enter and retrieve data. In a preferred embodiment, the user interface includes a pressure-sensitive display that allows a user to enter input with a stylus or other device. An example of a PID with such an interace is a PDA (Personal Digital Assistant), such as one of the Palm™ series of PDAs offered by 3Com® Corporation. The PIDs 110a-b may include other functionality, such as wireless phone or two-way radio functionality.

Links 109a-b are point-to-point links, and may be entirely or partially wireless, or they may be hard-wired connections. Each of the links 109a-b is preferably a wireless link, such as an infrared link specified by the Infrared Data Association (IrDA) (see irda.org for further information) or a radio frequency (RF) link such as the Bluetooth system (see www.bluetooth.com for further information). However, the point-to-point link can also be a hardwired connection, such as an RS-232 serial port.

In one embodiment, the data network telephone 108 and the data network teleports 18 et seq. include handsets with receivers and transmitters similar or identical to handsets of traditional circuit-switched telephones. A console on which the handset sits may include the voice processing system, a display 116, and a keypad 118.

In a preferred embodiment, a portion of the data network telephone 108 utilizes an NBX 100™ communication system phone offered by 3Com® Corporation. In alternative embodiments, the data network telephone 108 may include any device having voice communications capabilities. For example, a personal computer having a microphone input and speaker output may also be used to implement the data network telephone 108. Other configurations are also intended to be within the scope of the present invention.

The details relating to operation of the data network telephone 108 depend on the nature of the data network 106 and the nature of the access networks 112, 114 connecting the data network telephone 108 to data network teleports 18 et seq. and/or to other network entities. The access networks 112, 114 typically include any high bandwidth network adapted for data communications, i.e. a network having greater than 64,000 bits-per-second (bps) bandwidth. The access networks 112, 114 may link to the data network telephone 108 and to the telecommunications network access station 17 using an Ethernet LAN, a token ring LAN, a coaxial cable link (*e.g.* CATV adapted for digital communication), a digital subscriber line (DSL), twisted pair cable, fiberoptic cable, an integrated services digital network (ISDN) link, and wireless links. In embodiments that may not require bandwidth greater than 64,000 bps, the access networks 112, 114 may also include the PSTN and link the data network telephone 108 by an analog modem.

B. System for Connecting to Data Network Services Using a Data Network Telephony System

One advantage of the PID-Enabled Data Network Telephony System 100 in FIG. 1 is that it may be used to provide PID connectivity to the data network 106. In one embodiment, the PIDs 110a are able to connect to data network services through a user interface on the PID 110a. The PID 110a includes a web application for retrieving information that can be communicated from the Internet services 101 over the data network 106, transported across the first access network 112, to the telecommunications network access station 107 to the data network teleport 18a. The PID 110a can receive the information across the link 109a for display on the PID 110a.

In one embodiment, the PID 110a uses the Point-to-Point Protocol (PPP) to communicate with the data network teleport 18a. The PID 110a communicates requests

for services to the data network teleport 18a to send over the wireless connection 20 to the telecommunications network access station 107 connected to the data network 106. The PID 110a receives the Internet service offerings (*e.g.* web clippings™) from the data network through the wireless connection 20 between the telecommunications network access station 107 and the data network teleport 18a.

A voice-over-data channel for communicating voice-over-data may or may not concurrently exist with this communication of information over a data channel. In this way, a user of the PID 110a can receive information from the Internet services 101 while voice signals are communicated between the data network teleport 18a and the data network telephone 108b. Alternatively, the user may use the PID 110a connection to the Internet services independently of any telephone calls.

1. Local Area Network As An Exemplary Access Network

FIG. 2 is a block diagram showing one example of the system 100 of FIG. 1 for providing telephony services to multiple users in a residence and for accessing Internet services 101 using a PID 110 according to the present invention. The system 200 in FIG. 2 includes a local area network 212, connected to a data network 206 by a first router 228. A cable network 214 is connected to the data network 206 by a second router 238. The wireless residential telephone system 10 is connected to the cable network 214. Those of ordinary skill in the art will appreciate that while FIG. 2 illustrates the access networks as the local area network 212 and the cable network 214, any other type of network may be used. For example, the local area network 212 and/or the cable network 214 may be replaced by ISDN, DSL, or any other high-speed data link.

The local area network 212 provides data connectivity to its network elements, such as a first data network telephone 208a, a second data network telephone 208b, and a first network telephony connection server 150. The local area network 212 in FIG. 2 is an Ethernet LAN operating according to the IEEE 802.3 specification, which is incorporated by reference herein, however, any other type of local area network may be used. The local area network 212 uses the router 228 to provide the data network telephone 208a and the first network telephony connection server 150 with access to the data network 206. For example, the router 228 may perform routing functions using

protocol stacks that include the Internet Protocol and other protocols for communicating on the Internet.

The first network telephony connection server 150 provides telephony registration, location and session initiation services for voice connections in which its members are a party. A user may register for telephony service with an administrator of the first network telephony connection server 150 and receive a user identifier and a telephone identifier. The user identifier and telephone identifier may be sequences of unique alphanumeric elements that callers use to direct voice connections to the user. The first network telephony connection server 150 registers users by storing user records in a first registration database 152 in response to registration requests made by the user.

The call setup process and the user and telephone identifiers preferably conform to requirements defined in a call management protocol. The call management protocol is used to permit a caller anywhere on the data network to connect to the user identified by the user identifier in a data network telephone call. A data network telephone call includes a call setup process and a voice exchange process. The call setup process includes steps and message exchanges that a caller and callee perform to establish the telephone call. The actual exchange of voice signals is performed by a data communications channel. The data communications channel incorporates other data transport and data formatting protocols, and preferably includes well-known data communications channels typically established over the Internet.

The call management protocol used in FIG. 2 is the Session Initiation Protocol (SIP), which is described in M. Handley et al., "SIP: Session Initiation Protocol," IETF RFC 2543, Mar. 1999, incorporated by reference herein, however, any other such protocol may be used. Other protocols include H.323, the Media Gateway Control Protocol (MGCP), MEGACO, etc.

The network telephony connection server 150 may be used to provide telephony service for mobile users. A user may be registered to use the first network telephone 208a (which is identified by its telephone identifier), but move to a location near the second network telephone 208b. The user may re-register as the user of the second network telephone 208b. Calls that identify the user by the user's user identifier may reach the user at the second network telephone 208b.

2. Cable Network As An Exemplary Access Network

The system 200 in FIG. 2 also shows a cable network 214 connected to the data network 206 by a router 238. The cable network 214 provides data network access to its network elements, which in FIG. 2 include a third data network telephone 218a and a second network telephony connection server 162. The users of the data network telephone 218a connected to the cable network 214 may communicate over the data network 206 with the users of the data network telephones 208a-b connected to the local area network 212.

The cable network 214 includes any digital cable television system that provides data connectivity. In the cable network 214, data is communicated by radio frequency in a high-frequency coaxial cable. The cable network 214 may include a head-end, or a central termination system that permits management of the cable connections to the users.

3. Providing Telephony Services

The second network telephony connection server 162 is preferably a SIP-based server that performs call initiation, maintenance and teardown for the data network telephone 218a and for the data network teleports 18 et seq. in the wireless residential telephone system 10 connected to the cable network 214. The second network telephony connection server 162 may be similar or identical to the first network telephony connection server 150 connected to the local area network 212.

The system 200 shown in FIG. 2 permits the data network telephones 208a-b connected to the local area network 212 to communicate with the data network telephone 218a and with the data network teleports 18 et seq. in the wireless residential telephone system 10 connected to the cable network 214. The system shown in FIG. 2 uses SIP in order to establish, maintain, and teardown telephone calls between users.

There are two major architectural elements to SIP: the user agent (UA) and the network server. The UA resides at the SIP end stations, (e.g. the data network telephones), and contains two parts: a user agent client (UAC), which is responsible for issuing SIP requests, and a user agent server (UAS), which responds to such requests. There are three different network server types: a redirect server, a proxy server, and a registrar. The various network server types may be combined into a single server, such

as the network telephony connection server 150 and 162. Not all server types are required to implement the embodiments of the present invention. The communication services to be provided will determine which servers are present in the communication system. Preferred embodiments of the present invention may be carried out using proxy servers.

One example of a SIP operation involves a SIP UAC issuing a request, a SIP proxy server acting as end-user location discovery agent, and a SIP UAS accepting the call. A successful SIP invitation consists of two requests: INVITE followed by ACK. The INVITE message contains a user identifier to identify the callee, a caller user identifier to identify the caller, and a session description that informs the called party what type of media the caller can accept and where it wishes the media data to be sent. User identifiers in SIP requests are known as SIP addresses. SIP addresses are referred to as SIP Uniform Resource Locators (SIP-URLs), which are of the form *sip:user@host.domain*. Other addressing conventions may also be used.

Redirect servers process an INVITE message by sending back the SIP-URL where the callee is reachable. Proxy servers perform application layer routing of the SIP requests and responses. A proxy server can either be stateful or stateless. A stateful proxy holds information about the call during the entire time the call is up, while a stateless proxy processes a message without saving information contained in the message. Furthermore, proxies can be either forking or non-forking. A forking proxy can, for example, ring several data network telephones at once until somebody takes the call. Registrar servers are used to record the SIP address (the SIP URL) and the associated IP address. The most common use of a registrar server is for the UAC to notify the registrar where a particular SIP URL can be reached for a specified amount of time. When an INVITE request arrives for the SIP URL used in a REGISTER message, the proxy or redirect server forwards the request correctly.

At the local area network 212, the central registrar/proxy server, such as the first network telephony server 150, is the primary destination of all SIP messages trying to establish a connection with users on the local area network 212. Preferably, the first network telephony server 150 is also the only destination advertised to the SIP clients outside the LAN 212 on behalf of all the SIP clients residing on the LAN 212. The

network telephony server 150 relays all SIP INVITE messages to the appropriate final destination (or another SIP proxy), based on a database lookup using the first SIP database 152. It allows all mobile clients to register with their current locations.

Similarly, the second network telephony server 162 is the primary destination of all SIP messages trying to establish a connection with the data network telephone 218a and with the data network teleports 18 et seq. in the wireless residential telephone system 10 connected to the cable network 214. Preferably, the second network telephony server 162 is also the only destination advertised to the SIP clients outside the cable network 214 on behalf of all the SIP clients (*e.g.* data network telephones and data network teleports) residing on the cable network 214. The second network telephony server 162 relays all SIP INVITE messages to the appropriate final destination (or another SIP proxy), based on a database lookup using the second SIP database 164.

The data network telephones 208a-b, 218a and data network teleports 18a-c in the system 200 preferably have pre-programmed device identifiers (*e.g.* phone numbers), represented as SIP-URL's that are of the form *sip: user@domain*. An example is *sip: 8475551212@3Com.com.*. After power-up, each of the data network telephones 208a-b, 218a and the data network teleports 18a-c sends a SIP REGISTER message to the default registrar, such as the network telephony servers 150 and 162. The data network teleports 18a-c may have their SIP REGISTER messages sent by the telecommunications network access station 107.

When a call arrives at one of the network telephony servers 150 or 162 for any of the registered SIP URLs, the server will forward the call to the appropriate destination. If a data network telephone is moved to a new location, all calls to the associated SIP URL will still be properly routed to that device. In other words, the system in FIG. 2 provides device mobility in the sense that calls will "follow" the data network telephone according to its SIP URL. This is especially useful if the data network telephone 208a-b, 218a, or the telecommunications network access server 107 is running the DHCP (Dynamic Host Configuration Protocol) so that when the location is changed, the IP address is also automatically changed.

An advantage of the system in FIG. 2 is that once the call is established between data network telephones and/or data network teleports, the data network 206 provides

data connectivity for a plurality of data communications channels. For example, the data network telephones 208a, 218a, and the data network teleports 18 a-c can communicate voice signals as voice-over-data packets on a voice-over-data channel. The data network telephones 208a, 218a, and the data network teleports 18 a-c can also communicate data (such as PID data) as data packets on a data channel. For example, the data may be communicated to and from the PIDs 210a and/or 220a across links 209a and 219a to the data network telephones 208a and 218a, where data is packetized and depacketized as part of the process for communicating the data packets across the data network 206 and any access networks, such as the Ethernet LAN 212 and the cable network 214. The data channels may be established to communicate data to and from the Internet services 101.

4. The Data Network Telephones

The data network telephones 208a-b are preferably telephones that include an Ethernet communications interface for connection to an Ethernet port. The Ethernet phones in FIG. 2 support the Internet Protocol (IP), using an IP address that is either statically configured or obtained by access to a Dynamic Host Configuration Protocol (DHCP) server.

FIG. 3 is a block diagram showing the data network telephone 208a connected to the local area network 212 in FIG. 2. The data network telephone 208a in FIG. 3 is connected to the network 212 by a network interface 270. The network interface 270 may, for example, be a network interface card, and may be in the form of an integrated circuit. A bus 248 may be used to connect the network interface 270 with a processor 240 and a memory 242. Also connected to the processor are user interface circuitry 260 and three alternative link interfaces to a PID, such as the PID 210a.

A first link interface 248 includes an RS-232 serial connection and associated coupling hardware and mechanisms. The first alternative link interface 248 may, for example, be a docking cradle for a PDA (Personal Digital Assistant), in which information can be transferred between the PDA and the data network telephone 208a. The second alternative link interface comprises a first connection 254, such as an RS-232 connection, along with infrared circuitry 250 for converting signals into infrared output and for accepting infrared input. An infrared interface 252 may also be included within

the second alternative link interface. The third alternative link interface comprises a first connection 256, such as an RS-232 connection, along with radio-frequency circuitry 258 for converting signals into radio frequency output and for accepting radio frequency input. A radio frequency interface 259 may also be included as part of the third alternative link interface.

The three alternative link interfaces described above are merely examples, and additional means for implementing the link interface between the data network telephone 208a and the PID 210a may also be used. Although three link interfaces are shown in FIG. 3, there may be only one such interface in the data network telephone 208a. More than one link interface may be included to improve flexibility and to provide redundancy in case of failure of one of the link interfaces.

The user interface circuitry 260 includes hardware and software components that access the functions of the handset, display, and keypad to provide user input and output resources for functions in the processor 240. The user interface circuitry includes a display interface 262, a keypad interface 264, an audio output interface 265, and an audio input interface 267.

The audio input interface 267 may receive voice signals from a microphone or other audio input device and convert the signals to digital voice information. The conversion preferably conforms to the G.711 *ITU Standard*. Further processing of the digital signal may be performed in the audio input interface 267, such as providing compression (*e.g.* using G.723.1 standard) or providing noise reduction, although such processing may also be performed in the processor 240. Alternatively, the audio input interface 267 may communicate an analog voice signal to the processor 240 for conversion to digital information within the processor 240.

The audio output interface 265 receives digital information representing voice from the processor 240 and converts the information to audible sound, such as through a magnetic speaker. In one embodiment, the audio output interface 265 receives information in the form of G.711, although other processing such as decompression may be performed in the audio output interface 265. Alternatively, the processor 240 may convert digital information to analog voice signals and communicate the analog voice signals to the audio output interface 265.

The keypad interface 264 and the display interface 262 include well-known device interfaces and respective signal processing techniques. The user interface circuitry 260 may support other hardware and software interfaces. For example, a videophone implementation might also include a camera and monitor. The data network telephones of the present invention are not limited to telephones or videophones – additional user interface types, for example, such as the ones needed for computer games, are also contemplated as being within the scope of the present invention. In addition, some of the features described here, such as the display interface 262, are optional and serve to enhance the functionality of the first data network telephone 208a.

The processor 240 may consist of one or more smaller processing units, including, for example, a programmable digital signal processing engine. In the preferred embodiment, the processor is implemented as a single ASIC (Application Specific Integrated Circuit) to improve speed and to economize space. The processor 240 also may include an operating system, and application and communications software to implement the functions of the data network telephone 208a. The operating system may be any suitable commercially available embedded or disk-based operating system, or any proprietary operating system.

The processor 240 includes a media engine 241 and a signaling stack 243 to perform the primary communications and application functions of the data network telephone 208a. The purpose of the signaling stack in the exemplary data network telephone 208a is to set up, manage, and tear down a call. During the setup phase, a user may use the keypad to enter a user identifier to call. Alternatively, a PID such as PID 210a may transmit the user identifier of the party across the first link 209a. The signaling stack 243 receives the user entry and formats a request message to send to the user identified by the user identifier to initiate a telephone call. When the request message is sent, the location of the user identified by the user identifier is discovered, communication parameters, such as the supported voice CODEC types are exchanged, and a voice-over-data channel is established. During the management phase, for example, other parties may be invited to the call if needed. During the tear down phase, the call is terminated.

The signaling protocol used in the data network telephone 208a in FIG. 3 is the SIP protocol. In particular, the signaling stack implements a User Agent Client 244 and a User Agent Server 242, in accordance with the SIP protocol. Alternative signaling protocols, such as the ITU-T H.323 protocol, MGCP, MEGACO, and others, may also be used to implement the present invention.

Once the call is set up, the media engine 241 manages the communication over one or more data communications channels using network transport protocols and the network interface 270. The media engine 241 sends and receives data packets having a data payload for carrying data and an indication of the type of data is being transported. The media engine 241 in the data network telephones 208a may sample the voice signals from the audio input 267 (or receive voice samples from the audio input 267), encode the samples, and build data packets on the sending side. On the receiver side, in addition to performing the reverse operations, the media engine also typically manages a receiver buffer to compensate for network jitter. Similar procedures may be performed for other types of data, such as graphical data, or for data used in PID applications such as email, contacts data, calendar data, other non-voice sound data, interactive game data, etc.

The media engine 241 may also include hardware and software components for performing registration functions 247, voice-over-data functions 249, display data functions 251, and keypad output functions 253. The media engine 241 processes data that is received from the network 212, and data to be sent over the network 241.

For data that is received from the network 212, the media engine 241 may determine from the type of data in the packet (such as by examining a packet header) whether packets contain sampled voice signals or other data types. Packets containing sampled voice signals are processed by the voice-over-data function 249. The voice-over-data function 249 preferably conforms to a protocol for formatting voice signals as digital data streams. While any suitable protocol may be used, the media (i.e. the voice signal) is preferably transported via the Real Time Protocol (RTP), which itself is carried inside of UDP (User Datagram Protocol). RTP is described in H. Schulzrinne et al., "RTP: A Transport Protocol for Real-Time Applications," IETF RFC 1889, Jan. 1996, which is incorporated herein by reference. UDP is described in J. Postel, "User Datagram Protocol," IETF RFC 768, Aug. 1980, and IP is described in J. Postel, ed.,

"Internet Protocol," IETF RFC 791, Sept. 1981, both of which are incorporated by reference herein.

Packets containing data for use in registering the data network telephone 208a with a network telephony service are processed by the registration function 247. By registering the data network telephone 208a, a user may establish with the network telephony connection server 150 that calls addressed to the user's user identifier may be connected to the data network telephone 208a. Registration may occur when the data network telephone 208a sends a request to register to a service provider host, such as the network telephony connection server 150. The service provider host may respond by setting the user's user identifier to correspond to the telephone identifier of the data network telephone 208a, and by acknowledging the request with a status message to the data network telephone 208a. In one embodiment, a request to register the data network telephone 208a to a default user is automatically sent during power-up of the data network telephone 208a.

Other features may be added to the registration functions 247, or implemented as extensions to the registration functions 247. For example, the first data network telephone 208a may be provisioned to provide selected network telephony services by establishing a data connection with a service provider, requesting the selected services, and receiving data that ensures that the services have been successfully provisioned. Such services may include, for example, caller identification, call forwarding, voice mail and any other services offered by the network telephony service provider to enhance the capabilities of the first data network telephone 208a. One advantage of provisioning functions is that services may be ordered for temporary use in a manner convenient to the user.

Packets containing data for display on a display device of the data network telephone 208a are processed by the display data function 251. The display data function 251 may be used for displaying, for example, the names and user identifiers of other parties to the call, the status of the telephone call, billing information, and other information.

For data to be sent over the data network 212, the media engine 241 formats the data as data packets in accordance with a selected protocol. The selected protocol is

preferably a protocol that is supported by data network telephones that will receive the data being transported. The media engine 241 may include a data connection application 245 to perform functions relating to data connections over the data network 206. The data network telephone 208 may include a data connection management protocol (*e.g.* the hypertext transport protocol, or http) to handle data connections. Alternatively, the SIP protocol may be used to establish data connections as well as voice connections. The data connection application 245 may also perform proxy services to permit the PID 108 to establish data connections.

The voice-over-data function 249 formats voice samples according to the protocol used by the receiving data network telephone. In one preferred embodiment, the voice over data function 249 formats voice samples as RTP packets. The registration function 247 and the keypad output function 253 may control the transport of data that does not represent voice signals.

The data network telephones 208b and 218a are preferably similar or identical to the data network telephone 208a. For each of the data network telephones 208a-b and 218a, many of the features described in FIG. 3 are optional and their inclusion depends on the services to be offered.

5. The Telecommunications Network Access StationThe telecommunications network access station 107 provides the data network teleports 18 et seq. with access to the data network 206. The telecommunications network access station 107 may also serve as the host for telephone service to the wireless residential telephone system 10 in FIG. 1. During the registration or initiation of a telephone connection involving one of the data network teleports 18 et seq., the telecommunications network access station 107 may include a signaling stack 243' to perform the connection initiation steps described above for the data network telephone 208.

The telecommunications network access station 107 includes a wireless telephone interface 32 which performs the wireless connections to the data network teleports 18 et seq. when one or more of the data network teleports 18 et seq. are being used in a voice or data connection. The telecommunications network access station 107 includes a teleport channel 30 for each data network teleport 18 being used in a voice or data connection. The wireless telephone interface 32 may include a Direct Signal Spread

Spectrum interface at 2.4 Ghz, any spread spectrum (900 Mhz.) scheme, radio-frequency using Bluetooth or a similar specification. For example, the IEEE 802.11 Wireless Ethernet specification may also be used.

The telecommunications network access station 107 may include user data 24a such as the user identifiers and telephone identifiers in the memory 24. In a preferred embodiment, the telephone identifiers in the user data 24a may include port numbers in accordance with the User Datagram Protcocol (UDP) to identify the data network teleport that may be involved in a telephone or data connection. The port numbers are communicated as the source address in data packets transmitted to the other party or parties to the telephone connection.

Each data network teleport 18 may be identified using a SIP address, an Universal Resource Identifier (URI), or other type of identifier, during the use of the telephony service. The data network teleports 18 et seq. may be registered to the data network telephone service using a profile that includes the user's name, billing information, service information, user identifer and telephone identifier. The profile may be stored in the user's PID 110 (see FIG. 2) and transferred to the telecommunications network access station 107 using the link 109a to the data network teleport 18 and then to the telecommunications network access station 107.

The telecommunications network access station 107 may include a processor 240, media engine 241, signaling stack 243 and bus 239' similar to those used in the data network telephone 208 in FIG. 3A. The media engine 241 may also include the voice-over-data function 249, data connection application 245 and the registration function 247 similar to those in the media engine 241 in the data network telephone 208.

The voice-over-data function 249and the data connection application 245 perform data network communications functions for voice connections and data connections to the data network teleports 18 et seq. The voice-over-data function 249 communicates voice-over-data packets on at least one channel over the data network. The data connection application 245 communicates on at least one data channel over the data network. The telecommunications network access station 107 includes a teleport connection controller 50 to initiate connections between voice-over-data channels and corresponding teleport channels 30.

When a connection is initiated by one of the data network teleports 18 et seq., the teleport connection controller 50 determines which data network teleport 18 initiated the connection. The signaling stack 243 then initiates the session that comprises the telephone connection. The teleport connection controller 50 ensures that the resulting voice-over-data channel and/or data communications channel is coupled to the teleport channel 30 that corresponds to the data network teleport 18 that initiated the connection. In one embodiment, the teleport connection controller 50 communicates voice-over-data packets, to the data network teleport 18 by noting the UDP port address that is the destination of the packets. In this manner, the telecommunications network access station 107 is the only device that is given an IP address. The other devices (*i.e.* the teleports) are accessed by their UDP port numbers, which may be stored as, or with the telephone identifiers in the memory 24.

Similarly, when a remote device initiates the connection, the target teleport 18 et seq. is accessed via one of the teleport channels 30. The teleport 18 that is the target of the telephone connection is identified by the telecommunications network access station 107 by the UDP port number, or other telephone identifier.

7. The Data Network TeleportsFIG. 3C shows a block diagram of the data network teleport 18 in accordance with on embodiment of the present invention. The data network teleport 18 communicates with the telecommunications network access station 107 using wireless base interface 22. In a preferred embodiment, the wireless base interface 22 communicates using the 2.4 Ghz. Direct Sequence Spread Spectrum (DSSS) scheme. The data network teleport 18 includes many of the same functions and components include in the data network telephone 208 described above with reference to FIG. 3A. The signaling stack 243', however, is optional in the data network teleport 18. In addition, The data network teleport 18 may or may not communicate data packets with the telecommunications network access station 107.

In a preferred embodiment, the data network teleport 18 includes a port data connect function 26, a port registration function 28, a port voice-over-data function 25, a display data 251' and a key out functions 253'. The port data connect function 26 establishes one or more data communications channels through the wireless connection

20 to the telecommunications network access station 107 to the Internet services 101. The data communications channels may be opened, or created, for access by the PID over the PID interfaces 248', 250', 258'.

The port registration function 28 initiates the registration of the data network teleport 107 with the telephony server. The process may be started by uploading a profile from the PID via the PID interfaces 248', 250', 258'. The telecommunications network access station 107 performs the registration function through its signaling stack 243 and signals when the registration function is complete. The data network teleport 18 receives a user identifier 44a and is assigned a teleport number 44b to use in voice-over-data and data communications. In a preferred embodiment the teleport number 44b is a UDP port number.

The port voice-over-data function 25 may include the process of packetizing G.711 symbols in RTP packets for transmission to the telecommunications network access station 107. Alternatively, the function 25 may send the symbols for packetizing by the telecommunications network access station 107. The remaining functions operate similar to counterpart functions described above with reference to FIG. 3A.

8. The Portable Information Devices (PIDs)FIG. 4 is a block diagram showing the exemplary PID 210a that can communicate via the link 209a with the data network telephone 208a connected to the LAN 212. The PID 210a may be linked to the data network telephone 208a through a link interface 545. A bus 580 may be used to connect the point-to-point interface 545 with a processor 540, a memory 542, data storage 543, and user interface circuitry 544.

The link interface 545 shown in FIG. 4 illustrates three alternative link interfaces for establishing a link to a data network telephone, such as the data network telephone 208a.

A first link interface 546 includes an RS-232 serial connection and associated coupling hardware mechanisms. The first alternative link interface 546 may, for example, be for coupling with a PDA docking cradle, in which information can be transferred between the PDA and the data network telephone 208a. The second alternative link interface comprises a first connection 548, such as an RS-232 serial connection, along with infrared circuitry 250 for converting signals into infrared output

and for accepting infrared input. An infrared interface 552 may also be included within the second alternative link interface. The third alternative link interface comprises a first connection 554, such as an RS-232 connection, along with radio-frequency circuitry 556 for converting signals into radio frequency output and for accepting radio frequency input. A radio frequency interface 558 may also be included as part of the third alternative interface. The radio interface 554/556/558 may be implemented according to the Bluetooth specifications, described at www.bluetooth.com, or the IEEE 802.11 Wireless Ethernet specification. The three alternative link interfaces described above are merely exemplary, and additional means for implementing the interface between the PID 210a and the data network telephone 208a may also be utilized. Although three link interfaces are shown in FIG. 4, there may be only one such interface in the PID 210a. More than one link interface may be included to improve flexibility and to provide redundancy in case of failure of one of the link interfaces.

The user interface circuitry 544 includes hardware and software components that provide user input and output resources for functions in the processor 540. The user interface circuitry includes a display output 562, a display input 565, and an additional input/output interface 567.

The display output 562 preferably receives digital information representing graphical data from the processor 540 and converts the information to a graphical display, such as text and/or images, for display on a display screen, for example.

The display input 565 may receive data inputs, such as graphical data inputs, from a user of the PID 210a. The graphical data inputs are preferably entered by the user with a stylus on a pressure-sensitive display screen, and may include text, drawings, or other objects that are capable of being graphically presented.

The additional input/output interface 567 allows the user to enter other types of data besides graphical data into the PID 210a. For example, audio data, additional graphical data, or additional input, such as video camera input for example, may be entered through the additional input/output interface 567. The data may also include data formatted for operation with particular applications on the PID. For example, email data, calendar data, contacts data, database data, spreadsheets, notes, game data, etc. may also

be entered. Touch-sensitive screen buttons are an exemplary method for a user to enter control data into the PID 210a.

The processor 540 may include an operating system, as well as application and communication software, to implement the functions of the PID 210a. The operating system may be any suitable commercially available operating system, or any proprietary operating system. The operating system and software may be stored on data storage 543, in the memory 542, or the may be embedded in the processor 540. Although the processor 540 is shown connected to the data storage 543 through a bus 580, other configurations may also be used. Similarly, the memory 542 may be configured other than as shown in FIG. 4, and may be embedded within the processor 540.

The PID 210a is able to send data to and receive data from the data network telephone 208a across a point-to-point link, such as the point-to-point link 209a shown in FIG. 1. A user enters PID data at the display input 565. The graphical data may be processed in the user interface circuitry 544 or it may go directly to the processor 540 or the memory 542. The processor 540 may also perform processing functions, such as compression.

A PID data application may be used to perform functions that may implement the display input, the display output, and the processing functions. For example, a Web PID data service application 575 may be used to request and receive information from Internet services 101 (shown in FIG. 2) in a format suitable for the PID 210. The information or clippings are provided by the Internet services 101. The information retrieved as clippings could then be displayed through the display output 562 to enable the user to see a visual representation of the information.

If the user desires to request information from Web PID data service 103, a periodic request may be set up to make a request at a designated time as long as a link interface to the data network telephone 208 is active. Alternatively, the user may store links (*e.g.* hot links) and select the hotlinks using the stylus or other user input to request specific information at anytime. The request can be transmitted through one of the point-to-point interfaces 545, allowing the data to be received by the data network telephone 208a. An application in the data network telephone 208a receives the request across the point-to-point link, and the request is prepared for transmission across the data network

206, such as by the media engine 241 shown in FIG. 3. Preferably the request is converted to data packets and is communicated on a data channel across the LAN 212 through the router 228 across the data network 206 to the selected web clipping 103.

The web clipping 103 processes the request using well-known techniques (*e.g.* http). The requested information is formatted as data packets, preferably in the form of TCP/IP data packets to the data network telephone 208a. The data network telephone 208a may recognize the data packets as related to the previously made request and simply pass the information to the PID 210, or process the data packets.

The link 209a between PID 210a and the first data network telephone 208a can alternatively be implemented as an infrared link using all or parts of a specialized protocol, such as the Infrared Data Association (IrDA) protocol stack, where data is interpreted through the stack between application-layer processes at each end of the link.

FIG. 5 is a protocol diagram illustrating the layers of the IrDA protocol stack. An IrDA stack is implemented at each of the connection endpoints of an IrDA link. For example, the first PID 210a and the first data network telephone 208a could each implement an IrDA protocol stack to enable the link 209a. As a second alternative, two PIDs, such as the first PID 210a and the third PID 218a, may each contain an IrDA stack. In the second alternative, the communications between the PIDs and the data network telephones might take place without the assistance of IrDA. For example, IrDa data from the first PID 210a might be transmitted across the link 209a as a serial stream of data to the first data network telephone 208a, which might treat the IrDA data like any other data received from the first PID 210a. The first data network telephone 208a could then assemble the IrDA data into packets, such as TCP/IP packets for transport across the access and data networks to the third data network telephone 218a. The third data network telephone 218a may disassemble the packets and forward the IrDA data (without interpreting the IrDA portions) across the link 219a to the third PID 220a. The third PID 220a could then process the IrDA information received across the networks.

The required layers of an IrDA protocol stack are the physical layer 602, the IrLAP layer 604, the IRLMP layer 606 and the IAS layer 608. The physical layer 602 specifies optical characteristics of the link, encoding of data, and framing for various speeds. The IrLAP (Link Access Protocol) layer 604 establishes the basic reliable

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connection between the two ends of the link. The IrLMP (Link Management Protocol) layer 606 multiplexes services and applications on the IrLAP connection. The IAS (Information Access Service) layer 608 provides a directory or "yellow pages" of services on an IrDA device.

The IrDA protocol also specifies a number of optional protocol layers, these protocol layers being TinyTP 610, IrOBEX 612, IrCOMM 614 and IrLAN 616. TinyTP (Tiny Transport Protocol) 610 adds per-channel flow control to keep traffic over the IrDA link moving smoothly. This important function is required in many cases. IrOBEX (Infrared Object Exchange protocol) 612 provides for the easy transfer of files and other data objects between the IrDA devices at each end of the link. IrCOMM 614 is a serial and parallel port emulation that enables existing applications that use serial and parallel communications to use IrDA without change. IrLAN (Infrared Local Area Network) 616 enables walk-up infrared LAN access for laptops and other devices. The use of the optional layers depends upon the particular application in the IrDA device. The IrDA protocol stack is defined by such standards documents as “IrDA Serial Infrared Physical Layer Link Specification”, “IrDA ‘IrCOMM’: Serial and Parallel Port Emulation over IR (Wire Replacement)”, “IrDA Serial Infrared Link Access Protocol (IrLAP)”, “IrDA Infrared Link Management Protocol (IrLMP)”, and “IrDA ‘Tiny TP’: A Flow-Control Mechanism for use with IrLMP”, and related specifications published by the IrDA and available at http://www.irda.org/standards/specifications.asp and is incorporated by reference herein.

The IrDA protocol stack can be implemented at just the PID devices at the endpoints with the intermediate phones and networks simply providing a tunnel for the media stream attendant to the infrared links. Since PIDs, such as the Palm PDA, already have an IrDA stack implemented in them to support their infrared link to other devices and the benefits of the IrDA stack are already available. By using the layers of the IrDA protocol stack, the PID applications and the base applications in the phones can be simplified as the IrDA protocol layers take over certain functionalities. For example, the IrOBEX layer in each IrDA protocol stack can be used to transfer text and graphics object files, such as electronic business cards or whiteboard graphics, end-to-end between PID devices connected via data connected data network telephones.

With the IrDA stack being implemented only in the PIDs and not in the phones, only a small level of delay is introduced for stack interpretation by each PID and the connection provided is largely transparent to the applications in the PID devices, i.e. little or no modification to existing user applications in the PIDs is required. This approach may be more suitable for delay sensitive applications, such as interactive games involving the transfer of data between user applications in each PID.

It should be noted that the IrDA stack is written for a single infrared point-to-point interface and not for an infrared-to-network-to-infrared interface. As a result, the timers and retransmission schemes implemented in view of the single infrared point-to-point interface may not function properly for the extended network interface.

Alternatively, IrDA stacks can be implemented in the phones as well. By implementing IrDA stacks in the phones, the timing of the infrared interface is unaffected by a network delay. Also, additional functions and features can be implemented in the phones. For example, the phones can implement challenge and authentication where the phone requires the user, through the PID, to enter a password or other information to authenticate an authorized user. Similarly, the PID may also be used to transmit commands to the phone and receive status information via the IrDA stack. The approach taken will depend upon the requirements of the design and the particular application.

6. Providing Telephony and Access to Internet Services

FIG. 6 is a functional block diagram and protocol stack diagram illustrating an embodiment of the protocol stacks in the first PID 210a and the first data network telephone 208a that support link 209a. In the infrared RS-232 embodiment, the link interface circuitry 545 in the first PID 210a provides the physical layer 656, such as that specified by the Infrared Data Association (IrDA), that connects via link 209a to the link interface circuitry 260 implementing a physical layer 664 in the first data network telephone 208a. The data link layer 654 in the first PID 210a provides data link control for link 209a in transferring data to and from a PID application client 652. Similarly, the first data network telephone 208a includes a data link layer 662 and a base application server 600 that is configured to synchronize connection and other functions with the PID application 652 in the first PID 210a.

When PID 210a is activated, either through power-up or through a user input at the user interface 650, the synchronization application client 652 in the PID 210a may send the user's SIP URL across the link 209a to the first data network telephone 208a, where it is received by the synchronization application server 600. The synchronization application server 600 sends the SIP URL received from the PID 210a across connection 230 and the Ethernet LAN 212 through connection 243 to the network telephony connection server 150. The network telephony connection server 150 may store the SIP URL and the IP address of the associated data network telephone 208a in the SIP database 152 so that the SIP URL is listed as being resident at the IP address of the data network telephone 208a. (If the network telephony connection server 150 uses a location server for registration/location tasks, the registration information might instead be stored with such a location server). SQL (Structured Query Language) is preferred for querying the database. Once the PID 210a is registered with the network telephony connection server 150, calls to the SIP URL for PID 210a (or the user of the PID 210a) will be directed to the data network telephone 208a.

FIG. 7A is a functional block and protocol stack diagram illustrating an embodiment of the present invention where a SIP connection is established from the first data network phone 208a to the third data network phone 218a through network connection 230, first access network 212, data network 206, second access network 214 and network connection 236. The routers 228 and 238, and associated connections 232a-b and 234a-b, are not shown to simplify the block diagram representation.

The diagram of FIG. 7A shows how requests for Internet services can be transmitted and responses to the requests processed in one aspect of the present invention. The PID application 652 in PID 210a is configured to send PID data as input, which in the present context is a request for data, such as a hotlink, or an URL. The request is sent through the user interface 650 through link 209a to base application 660 in the first data network phone 208a. In this embodiment, base application 660 is configured to define data channels for transport to the Internet services 101. The Internet services 101 responds to the request by sending back requested services to the PID 110a. For example, the Internet services may send back data from Web PID data service 686 (using WAP) to the PID 110a as PID data.

Multiple data channels in SIP may be defined through the Session Description Protocol described in RFC 2327, herein incorporated by reference. Included in a SIP INVITE request for a connection are options for the requested connection that describe the number and type of media streams. Each media stream is described by a "m=" line in the INVITE request. For example, a request for a connection that includes an audio stream and a bidirectional video stream using H.261 might look like this:

```
v=0
o=alice 2890844526 2890844526 IN IP4 host.anywhere.com
c=IN IP4 host.anywhere.com
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
m=video 51372 RTP/AVP 31
a=rtpmap:31 H261/90000
```

TABLE 1.

If the called device includes functionality to receive the connection as described in Table 1, then the called device will respond to the INVITE request with a 200 OK response that includes the same option values. If the called device or party is unable or unwilling to receive such a connection, then it will respond with alternative option values for the connection. See RFC 2543 for further details regarding the negotiation of connection parameters in SIP.

In FIG. 7A, a first data channel for voice data has been negotiated by the base applications 660 in the first data network telephone 208a and the base application 674 in the second data network telephone 218a. In addition, a second data channel for Internet services data has been negotiated by the base application 660 in the first data network telephone 208a and Web PID data service application 686 in the Internet services 101.

The base applications 660 and 674 transfer voice data between the AUDIO applications, such as applications including G.711 encoders, in each phone via the first data channel. The base application 660 in phone 208a is also configured to send data requests received via link 209a from PID 210a to the Internet services 101 via the second

data channel. The Internet services 101 processes the request and provides the requested information over the second data channel.

One advantage of the embodiment in FIG. 7A is that the voice channel is optional. The users may request the data connection to the Internet services 101 while conversing on the phones 208. Alternatively, a user may connect to the Internet services 101 independent of any voice connection.

FIG. 7B shows an alternative embodiment for providing a data connection from the PID 110a to the Internet services 101. The link 209a in FIG. 7B is an RS232 connection. The PID 110a includes an IP stack that includes a Point-To-Point client 653. In addition, the telephone 208a includes a PPP server 663. The PID 110a may connect directly to the Internet services 101 with its own IP address. One advantage of using the PPP-based connection in FIG. 7B is that even a basic, low-cost PID 110a may perform sophisticated Internet communications because the PPP is widely available at a low cost.

FIG. 7C shows a functional block diagram and protocol stack layers for providing telephony and data connections between the data network teleport 18, the PID 110, the telecommunications network access station 107 and either, another voice communications device 218, or an Internet service 102.

The PID 110 may communicate user input at a user interface 652 to a PID application 652 that communicates with the Internet services 102. The PID data is communicated on the PID link interface 109 to the data network teleport 18. The data network teleport 18 receives the PID data and communicates it to a port communications application 60, which may include the data connect application 26 shown in FIG. 3C. The port communications application 60 may also include the voice-over-data application 25 in FIG. 3C which may receive or communicate voice signals over the audio 658. The voice signals, as G.711 symbols, for example, and/or the PID data may be packetized and communicated via the port voice and data function 62 and the DSSS function 66 on a wireless connection 20 to the DSSS interface 64 on the telecommunications network access station 107.

The DSSS 64 communicates the signals to the network access voice and data function 68. A communications application in the network access station 107 communications the voice and data to a base application 660. The base application 660

may include the voice-over data function 249 and data connect function 245, shown in FIG. 3B. The data is then communicated via the HTTP 667 while the voice is communicated via the SIP 666 connections as described above with reference to FIG. 7A.

C. Accessing Internet Services Concurrent with Voice Services

FIG. 8 shows an exemplary embodiment of the present invention for transmitting data from an Internet service concurrently with voice services during a telephone conversation. The PID 210a includes a display screen 702, a stylus 700 that a user can use to select a hotlink, or URL address to a Web service. Alternatively, a SYNC button 718 may be used to initiate a series of requests to the Internet services 101.

The display screen 702 is shown as a pressure-sensitive display screen in which the stylus 700 can be used to enter PID data 714 into the first PID 210a. In the example shown in FIG. 8, the PID 210a includes hotlink 706 for accessing prices of selected stocks from the Internet services 101. The stylus is being used to select the hotlink 706 the stock prices.

In the embodiment shown in FIG. 8, the hotlink 706 is transmitted across the link 209a to the first data network telephone 208a. When the first data network telephone 208a receives the transmitted hotlink 714, an application within the first data network telephone 208a will place the hotlink 714 into PID data packets for transmission to the Internet services 101 across the access and data networks 212, 206 (and any associated connections and routers). The Internet services then processes the hotlink 714 and responds by sending the requested stock prices in the PID data channel 724 back to the data network telephone 208. The data network telephone 208 transmits the information to the PID 208 for display on the PID display 702.

While the invention has been described in conjunction with presently preferred embodiments of the invention, persons of skill in the art will appreciate that variations may be made without departure from the scope and spirit of the invention. For example, the access networks shown in FIG. 2 may comprise any other suitable type of local area network or service infrastructure.

In addition, protocols of various types are referenced throughout. While preferred and alternative embodiments may implement selected protocols, any suitable replacement

protocol not mentioned, or any function not part of a protocol used to replace a corresponding function from a protocol may be implemented without departing from the scope of the invention.

This true scope and spirit is defined by the appended claims, interpreted in light of the foregoing.